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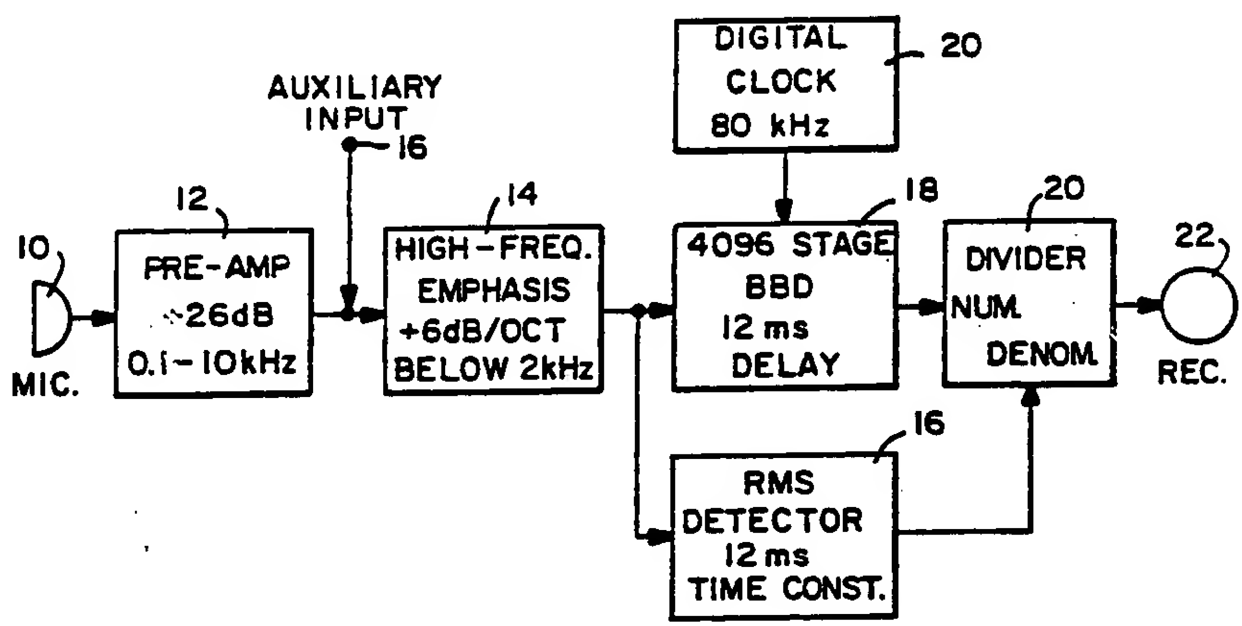
(54) Title: METHOD AND MEANS FOR PROCESSING SPEECH



(57) Abstract

A method of and apparatus for processing audio signals in which a measure of amplitude of audio signals in a selected time period is obtained. The audio signals (Fig. 1) for the selected time period are delayed (18) until the measure of amplitude (16) is obtained, and then the delayed audio signals are normalized (20) using the measure of amplitude. High frequency emphasis (14) may be employed prior to obtaining the measure of amplitude. Alternatively, a multi-channel system (Fig. 3) can be employed for processing audio signals in limited frequency bands (32, 34, 36). The method and apparatus are applicable in a variety of applications including hearing aids, audio storage media, broadcast and public address systems, and voice communications such as telephone systems.

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METHOD AND MEANS FOR PROCESSING SPEECH

This invention relates generally to speech processing, and more particularly the invention relates to a method and means for amplifying speech, such as for the hard of hearing, without adversely affecting the signal intelligence thereof.

It is well recognized that persons having sensorineural hearing impairment generally have a very limited dynamic range, that is, very little difference between the intensity level of the softest speech of which they are aware (e.g. speech awareness threshold or SAT), the intensity level of speech which is most comfortable for them (most comfortable level or MCL), and the intensity at which speech becomes too loud to be tolerated (uncomfortable loudness level or UCL). It is generally agreed that it would be highly desirable to reduce the wide range of speech intensity levels to a more restricted range suitable for the sensorineural hearing impairment of each individual listener.

Speech compression systems are known which employ automatic gain control. However, prior art systems employing peak clipping and instantaneous compression produce harmonic distortion which tends to emphasize the stronger, low-frequency components of speech and obscures the higher frequencies. A comprehensive

survey is presented by Braida et al in "Hearing Aids - A Review of Past Research on Linear Amplification, Amplitude Compression, and Frequency Lowering", American Speech-Language-Hearing Association, Rockville, Maryland, April 1979. This survey provides an extensive critical review of the compression literature in conjunction with a tutorial on compression concepts. The survey suggests that the lack of benefits from compression as shown in the survey literature reflects more a failure of researchers to adequately grasp the concepts and complexity of compression, in theory and implementation, rather than the potential benefit of amplitude compression itself.

It is recognized that the acoustical patterns of speech can be systemically analyzed in three primary time-domain components: (1) a fine-temporal pattern reflecting the spectral distribution of each brief acoustic segment, (2) a gross-temporal pattern reflecting the durations of the various acoustic segments based on changes in fine-temporal patterns, and (3) a time-varying amplitude pattern. The fine temporal cues from segments of speech as short as five or ten milliseconds will often provide a listener with sufficient information to identify the place of articulation for consonants. Similarly, the gross temporal pattern will often provide sufficient information regarding the manner of articulation, especially among the classes of fricatives, affricates and stop-plosives. The time varying amplitude pattern, or "speech envelope", is the natural result of a speech production process but may convey mostly redundant information already conveyed by a gross-temporal pattern. Robinson and Huntington, in a talk before the Acoustical Society of America in April, 1973, recognized that conventional compression amplification introduces undesirable distortion when brief time constants are utilized, and reacts too sluggishly for longer time constants. A

method was proposed in which the average power of the speech wave form over intervals of several tens of milliseconds is measured continuously and is used to determine the gain to be applied to the waveform at the center of each interval, with the resulting amplitude compressed signal being delayed by one-half the length of the averaging interval. Preliminary results from a computer simulation suggested that speech intelligibility could be improved by this process. However, further work was not undertaken by Robinson and Huntington to develop the process.

An object of the present invention is an improved method of processing speech to facilitate reception without distorting the intelligible content thereof.

Another object of the invention is apparatus for compressing speech patterns whereby the variations in time varying amplitude pattern or envelope are minimized without adversely affecting the fine-temporal and gross-temporal patterns of the speech.

The present invention is directed to a method and apparatus for processing speech in which a time-varying averaged or root-mean-square (RMS) amplitude pattern is obtained and used to normalize the time varying amplitude pattern of speech and provide a compressed speech pattern positioned between the speech awareness threshold and the uncomfortable loudness level, ideally at the listener's most comfortable level. Spectral shaping is employed to emphasize the high-frequency content. The invention can be implemented in a single channel or multi-channel system. Suitable microphone means is employed to pick up a speech pattern, and the speech pattern from the microphone is preamplified and then processed by a suitable shaping filter which emphasizes the high frequency content thereof. The root-mean-square of

the amplitude of the spectrally shaped signal is then determined over a specific time period, and the inverse of the root-mean-square is then used to modulate the spectrally shaped signal, thus producing a normalized amplitude. Importantly, the shaped signal is delayed for a sufficient time period to compensate for the time delay involved in the root-mean-square determination prior to the amplitude compression. The resulting signal is thus compressed and then adjusted to the desired hearing range with the spectral shaping providing a retention of the fine-temporal pattern and the gross-temporal pattern.

The invention and objects and features thereof will be more readily apparent from the following detailed description and appended claims when taken with the drawing, in which:

Figure 1 is a functional block diagram of a single channel speech processing apparatus in accordance with one embodiment of the present invention.

Figure 2 is a graph illustrating the compression of speech in accordance with the present invention.

Figure 3 is a functional block diagram of a multi-channel embodiment of speech processing apparatus in accordance with the invention.

Figures 4A and 4B are functional block diagrams of a tape recording system in accordance with other embodiments of the invention.

Referring now to the drawings, Figure 1 is a functional block diagram of a single channel speech processor in accordance with one embodiment of the invention which has been built using conventional, commercially available components. In this embodiment a microphone 10

having a broad frequency response (e.g. an electret microphone having a response of 100 Hertz to 10K Hertz such as a Knowles EA 1934) picks up audio signals and transmits electrical signals to a pre-amplifier 12 having 26 dB of gain between 100 Hertz to 10K Hertz. The amplified signal is then passed to high frequency emphasis circuity 14 (e.g. TI064 quad amplifier) which provides 6 dB/octave gain over the range from 100 Hertz to two kiloHertz and a flat response above two kiloHertz. An auxiliary input is provided at 16 whereby signals from a radio receiver, for example, can be applied to the high frequency emphasis circuitry 14.

The signal from circuitry 14 is then passed to an RMS detector high-frequency emphasis circuitry 14 is also provided to delay circuitry 18 having a delay equal to the time constant of the RMS detector 16. In one embodiment the RMS detector comprised an analog series AD 536A and the delay circuitry 18 comprised a Reticon SAD 4096 bucket brigade device operated from a 80 kilohertz digital clock 20.

The delayed signal from the delay device 18 is then applied as the numerator in a divider circuit 20 (e.g. Analog Devices AD 535 precision divider) and the RMS amplitude of the delayed signal is applied to the divider 20 as a denominator. Accordingly, the output from the divider 20 is a delayed amplitude compressed signal which is applied to the receiver 22 (Knowles ED 1925).

Figure 2 is a plot of the compressed output level in dB SPL for the signal applied to receiver 22 versus the input level in dB SPL of the signal from the microphone 10. For input levels below about 45 dB, the output level is attenuated. At an input level of 45 dB, the output level is compressed and maintained

uniform at approximately 100 dB SPL which is the MCL level. The compression ratio remains at 10:1 or greater for input levels above 45 dB.

Figure 3 is a multi-channel signal compression system in accordance with another embodiment of the invention in which signals are filtered and compressed in a plurality of frequency bands. In this embodiment signals from the microphone 30 are applied to a low band (100-400 hZ) filter 32, a middle band (400-1,600 Hz) filter 34, and a high band (1,600-6,400Hz) filter 36. Signals from each of the filters are passed to amplitude compressor circuitry 38, 40, and 42. Each of the compressor circuits includes delay circuitry, RMS detector circuitry, and divider circuitry as illustrated in Figure 1. Because each channel includes a narrow band of frequencies, the high frequency emphasis circuitry of Figure 1 is not required. The compressed signals are then applied to a summing amplifier 44 with the composite summed signal then applied to the receiver 46.

Figures 4A and 4B are functional block diagrams of other embodiments of the invention useful with tape recorders and in which the compressed signal and the detected RMS value are both recorded in time sequence with a tape recorder. In Figure 4A signals from the microphone 50 or other audio source are applied to amplitude compressor 52 which may be a single channel device as in Figure 1 or a multiple channel device as in Figure 3. The compressed audio signal is then recorded in an analog channel of the tape recorder 54, and the detected RMS value is recorded in an FM channel of the recorder 54. Thereafter, the recorded compressed audio signal and the recorded RMS value can be applied to a multiplier 56 from which the original audio signal and the original dynamic range is produced. The resulting decompressed signal is applied through

frequency de-emphasis circuit 58 to the receiver 59.

Figure 4B is a sampled digital recording system similar to the analog system of Figure 4A. In this embodiment signals from microphone 60 are applied to the amplitude compressor 62, as in Figure 4A, and then the compressed audio signal and the RMS value are converted to digital form by analog to digital circuits 63 and 65. The digital signals are then stored in digital recorder 64. The recorded signals are converted back to analog signals by digital to analog converter 57 and multiplying DAC 66. The decompressed signals from DAC 66 are frequency de-emphasized at 68 and then applied to the receiver 69.

These embodiments of the invention are particularly advantageous since tape recorders typically have a limited dynamic range. Thus, by recording the compressed audio signal and the RMS on the recorder, the full dynamic range of the recorded signal can be reconstructed in the multiplier 56 and multiplier 66.

In the preferred embodiments described herein, an RMS detector has been employed. However, other measures of the signal amplitude over a period of time, including an average value and an approximation of the RMS value, can be employed. As used herein, RMS value includes suitable approximations thereof. Further, while a divider has been employed in the preferred embodiments for obtaining the compressed signal, a logarithmic measure of the detected RMS or averaged value can be employed for obtaining the compressed value.

The invention has broad applications including, for example, hearing aids and audio storage media (as described herein), sampled digital storage system, broadcast systems, public address systems, and general

voice communication including telephones. The invention is especially useful for communication in a noisy environment and through a noisy communication link such as in field applications.

Thus, while the invention has been described with reference to specific embodiments, the description is illustrative of the invention and is not to be construed as limiting the invention. Various modifications and applications may occur to those skilled in the art without departing from the true spirit and scope of the invention as defined by the appended claims.

WHAT IS CLAIMED IS:

1. Apparatus for processing audio signals including speech comprising means for obtaining a measure of amplitude of speech signals during a selected period of time, and means for compressing audio signals during said selected period of time based on said measure of amplitude.

2. Apparatus as defined by Claim 1 wherein said means for obtaining a measure of amplitude comprises a root-mean-square (RMS) detector.

3. Apparatus as defined by Claim 2 and further including means for delaying audio signals for said selected period of time while said root-mean-square detector obtains the root-mean-square value of amplitude of said audio signals for said period of time.

4. Apparatus as defined by Claim 1 and including delay means for delaying audio signals for said selected period of time while said measure of amplitude of audio signals during the selected period of time is obtained.

5. Apparatus for processing audio comprising

microphone means for receiving audio and generating electrical signals in response thereto,

high frequency emphasis means connected to said microphone means for amplifying said electrical signals,

amplitude detection means connected to said high frequency emphasis means for receiving amplified signals and obtaining a root mean square (RMS) amplitude of said amplified signals over a selected period of time,

delay means connected to said high-frequency emphasis means for receiving and delaying spectrally

shaped electrical signals for said selected period of time, and

signal compression means connected to said delay means and to said amplitude detection means and compressing the delayed amplified electrical signals in accordance with said root-mean-square average value.

6. Apparatus as defined by Claim 5 wherein said high frequency emphasis means amplifies said electrical signals by six dB per octave at least to 2 kiloHertz.

7. Apparatus as defined by Claim 5 wherein said delay means comprises a bucket brigade device.

8. Apparatus as defined by Claim 5 wherein said compression means comprises a divider for dividing said delayed non-linearly amplified signals by said root-mean-square value.

9. Apparatus as defined by Claim 5 and further including pre-amplification means interconnecting said microphone means to said high-frequency emphasis means.

10. In audio signal processing apparatus, a method of compressing the amplitude of signals comprising the steps of

obtaining a measure of amplitude of said signals over a selected time period, and

compressing said signals corresponding to said selected period of time in accordance with said measure of amplitude.

11. The method as defined by Claim 10 wherein said measure of amplitude is a root-mean-square value.

12. The method as defined by Claim 11 wherein said step of compressing comprises dividing said signals for a selected period of time by the root-mean-square

amplitude value thereof.

13. The method as defined by Claim 10 and further including the step of emphasizing high frequency components of said signals prior to obtaining said root-mean-square value.

14. Apparatus for processing audio signals comprising

a plurality of band pass filters for receiving and filtering audio signals into a plurality of limited frequency bands,

a plurality of amplitude compressor means each connected with a band pass filter, each of said amplitude compressor means including

means for obtaining a measure of amplitude of audio signals during a selected period of time,

means for delaying audio signals for said selected period of time, and

means for compressing the delayed audio signals during said period of time based on said measure of amplitude, and

summing means connected to said plurality of amplitude compressor means for receiving and summing compression of audio signals.

15. Apparatus as defined by Claim 14 wherein said means for obtaining a measure of amplitude comprises a root-mean-square (RMS) detector.

16. A method of recording audio signals and obtaining original dynamic range from the recorded signals, said method comprising the steps of

obtaining a measure of amplitude of audio signals over a selected time period,

compressing audio signals corresponding to said selected time period in accordance with said measure

of amplitude,

recording the compressed audio signals, and

recording said measure of amplitude of audio signals.

17. The method as defined by Claim 16 wherein said measure of amplitude is a root-mean-square value.

18. The method as defined by Claim 16 and further including the steps of

retrieving the recorded compressed audio signals and the recorded measure of audio signals, and

decompressing the recorded compressed audio signals using the recorded measure of amplitude of audio signals.

19. The method as defined by Claim 18 wherein said step of recording includes converting compressed analog signals and measure of amplitude to digital form and said step of decompressing includes converting recorded digital signals to analog form.

20. The method as defined by Claim 16 wherein said step of recording includes converting compressed analog signals and measure of amplitude to digital form.

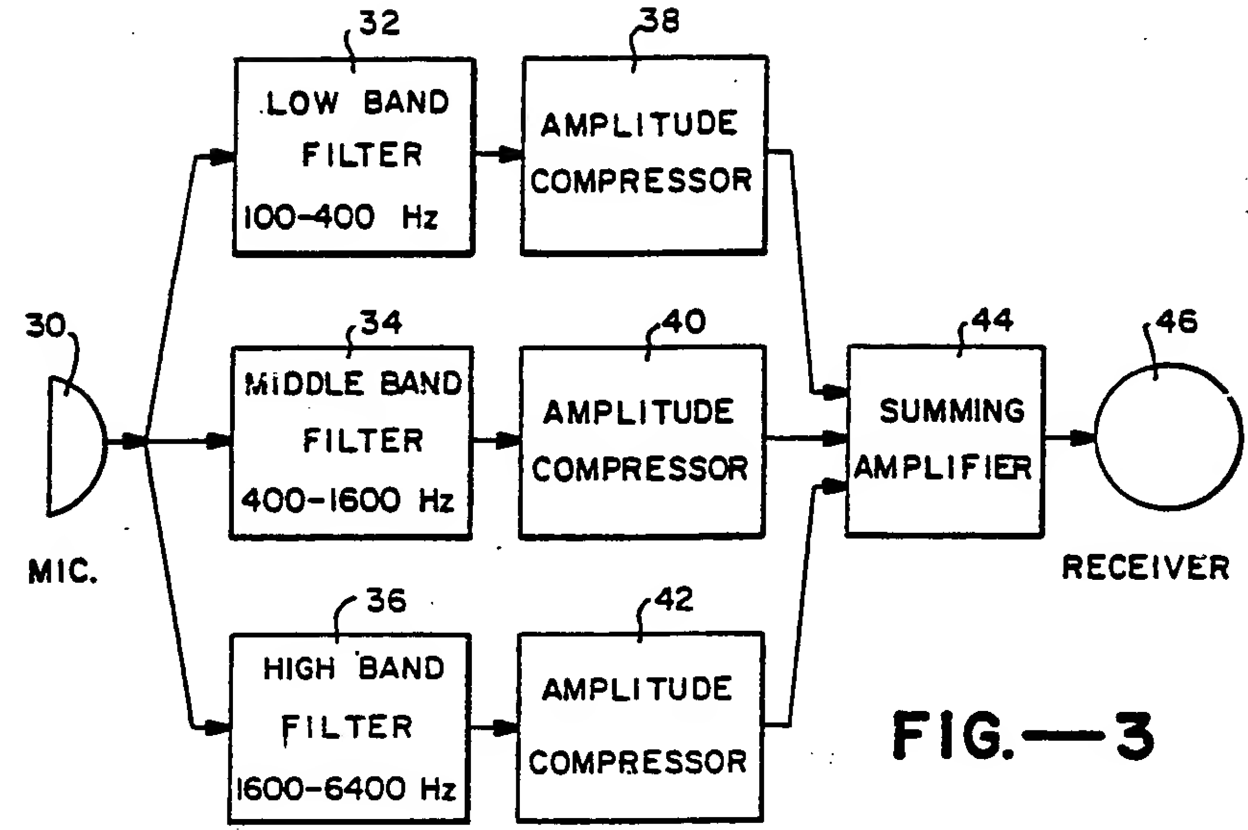

FIG.—3

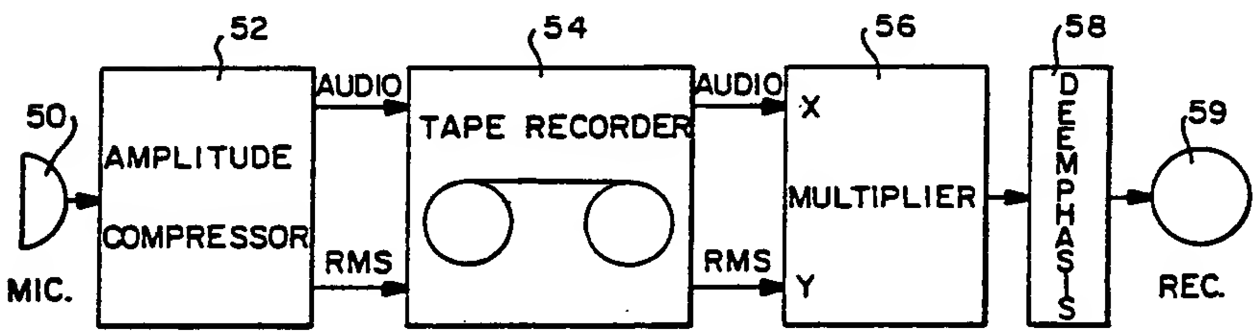

FIG.—4A

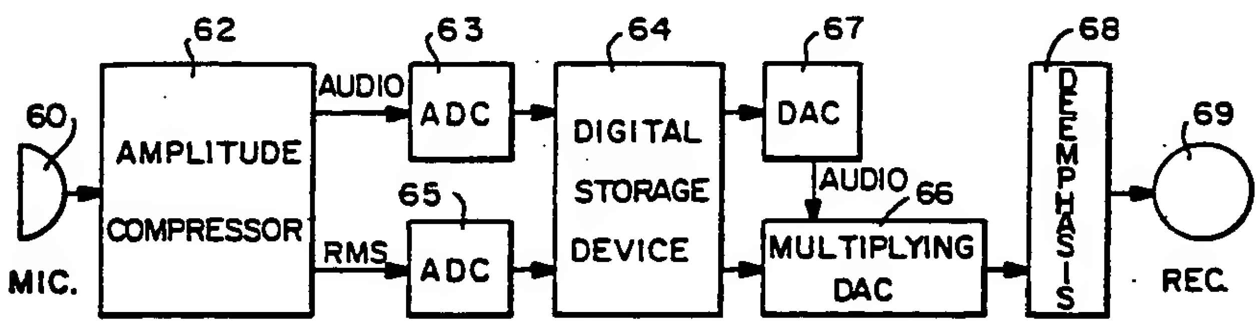

FIG.—4B

SUBSTITUTE SHEET

INTERNATIONAL SEARCH REPORT

International Application No PCT/US83/02030

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all) [3]

According to International Patent Classification (IPC) or to both National Classification and IPC

US CL. 381-106
INT. CL 3: G10L 1/00; H04B 1/66

II. FIELDS SEARCHED

Minimum Documentation Searched [4]

Classification System	Classification Symbols
US	381-106, 110; 179-107.R, 107.FD

Documentation Searched other than Minimum Documentation to the Extent that such Documents are Included in the Fields Searched [5]

III. DOCUMENTS CONSIDERED TO BE RELEVANT [14]

Category *	Citation of Document, [16] with indication, where appropriate, of the relevant passages [17]	Relevant to Claim No. [18]
Y	US, A, 4,112,254, 05 September 1978, Blackmer	1-20
Y	US, A, 4,249,042, 03 February 1981, Orban	1-20
Y	Journal of Acoustical Society of America, Vol. 54, No. 1, issued 1978, USA, C.E. Robinson et al, "The Intelligibility of Speech...", see page 314.	1-20

* Special categories of cited documents: [15]
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"&" document member of the same patent family

IV. CERTIFICATION

Date of the Actual Completion of the International Search [2]	Date of Mailing of this International Search Report [2]
08 February 1984	15 FEB 1984
International Searching Authority [1]	**Signature of Authorized Officer [20]**
ISA/US	E S Kemeny E. S. Kemeny

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